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(54) Transmitting a voice signal and data on a single channel.

(57) A method of and apparatus for the simultaneous transmission of voice and data information over the same PCM transmission channel. During periods when only voice information is to be transmitted, the analog voice information is sampled at a first sampling rate, thus providing a digitized voice rate equal to the transmission rate capability of the transmission channel. During periods when both voice and data are to be transmitted, the analog voice information is sampled at a second sampling rate less than the first sampling rate, thus allowing the merged voice and data information to have a total digitized transmission rate equal to the transmission rate capability of the transmission channel.

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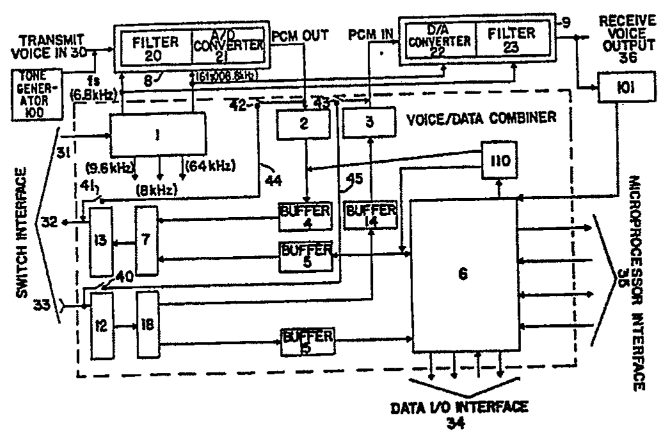

TRANSMIT VOICE IN 30
FILTER 20
A/D CONVERTER 21
PCM OUT
PCM IN
D/A CONVERTER 22
FILTER 23
9
RECEIVE VOICE OUTPUT 36
TONE GENERATOR 100
fs (6.8kHz)
8
101
42
43
1
2
3
VOICE/DATA COMBINER
31
(9.6kHz)
(8kHz)
(64kHz)
44
45
110
41
SWITCH INTERFACE
32
13
7
BUFFER 4
BUFFER 14
BUFFER 5
6
40
33
12
18
BUFFER 15
MICROPROCESSOR INTERFACE 35
DATA I/O INTERFACE 34

FIG. 1

TRANSMITTING A VOICE SIGNAL AND DATA ON A SINGLE CHANNEL

This invention relates to a method of and apparatus for transmitting a voice signal and data on a single channel.

Transmission systems employing pulse code modulation (PCM) are well-known. For example, there are telephone communication systems which employ pulse code modulation, in which an analog voice signal is converted to a series of binary pulses. Such telephone communication systems use a fixed sampling rate, commonly 8 kilohertz. By converting the sampled speech signal to an 8 bit word, the digitized speech is transmitted by pulse code modulation at 64 kilobits per second. Prior art speech transmission employs a fixed sampling rate because prior art filters use RC circuits to provide the filter characteristics. By the use of fixed resistors and capacitors, the characteristics of prior art filters are also fixed.

Data transmission, such as from a computer or microprocessor, also utilizes a series of binary pulses. One common data transmission rate is 9.6 kilobits per second.

Prior art methods of transmitting both speech and data require either separate voice and data transmission channels, or a single channel of increased bandwidth capable of carrying both data and speech. Thus, for a voice transmission of 64 kilobits per

second to be combined with a data transmission of 9.6 kilobits per second, a transmission channel capable of carrying 73.6 kilobits per second is required. In many instances this is not practical or possible. For example, standard PCM channels and equipment are designed to be capable of handling 64 kilbits per second, and it is not possible to transmit information in excess of this rate without redesigning the receiving equipment.

The invention accordingly provides a method of transmitting a voice signal and data information on a single channel, the method comprising the steps of

(a) sampling the voice signal at a first rate in the absence of data to be transmitted, and

(b) sampling the voice si nal at a second rate, lower than the first rate, and sampling the data information at a third rate lower than the first rate, during the presence of data to be transmitted.

Thus, a single voice channel capable of carrying 64 kilobits per second can be used to carry either voice information, data information, or a combination of voice information and data information. A sampling rate of eight (8) kilohertz can be used to provide a pulse code modulated voice signal of 64 kilobits per second during the interval when voice information is transmitted while during the periods when a combination of voice information and data information)or data information alone) is transmitted, a voice sampling rate of 6.8 kilohertz is used, resulting in a digitized voice rate of 54.4 kilobits per second. This 54.4 kilobits per second voice rate allows the addition of data information, at a rate of 9.6 kilobits per second, thus allowing simultaneous voice inforamtion and data information transmission over a single 64 kilobits per

second pulse code modulation voice channel. The reduction in voice bandwidth from 3.4 kilohertz to 2.9 kilohertz during data transmission periods does not degrade speech quality significantly, thus allowing transmission of high quality speech and data information simultaneously over a standard 64 Kilobit/sec telephone system.

The invention is further described below by way of example with reference to the accompanying drawings, in which:

Figure 1 is a block diagram of the circuit of a transmission system embodying this invention;

Figure 2a graphically represents the relationship of data and voice information to a single transmission frame;

Figure 2b represents a synchronizing signal for use in the invention;

Figure 2c further represents the relationship of data and voice information to a single transmission frame;

Figure 2d represents the relationship of a single data byte and a single voice byte to their component bits;

Figure 2e represents pulse code modulated voice information; and

Figure 2f represents continuous serial data information.

The transmission system shown in Figure 1 is capable of allowing simultaneous transmission of data information and voice information. During the periods of "full voice" operation, when voice information is transmitted without the simultaneous transmission of data information, a voice signal is input via a lead 30 to a pulse code modulator or "encoder" 8 which comprises a filter 20 and an analog to digital converter 21. Device number S3501 manufactured by American Microsystems, Inc. may be used as the encoder 8. The pulse code modulation

output from the encoder 8 is connected to an output lead 32, which is connected to a switching network via a PCM transmission channel (not shown). Switches 41 and 42 allow this pulse code modulated output from the encoder 8 to be directly applied to the output lead 32, therefore bypassing various components contained within a voice/data combiner. The output lead 32 is connected to the PCM channel (not shown).

Also, during the full voice operation, a receiver lead 33 is connected via a switch 40 and a switch 43 to a pulse code modulation decoder 9 comprising a digital to analog converter 22 and a filter 23. The decoder 9 may comprise an S3502 device also manufactured by American Microsystems, Inc. The output from the decoder 9 is connected to a suitable transducer device (not shown) to provide an audible output for human use. The sampling rate for encoding and decoding the voice signal during the full-voice mode is preferably 8 kilohertz, thus allowing the transmission of 8000 8-bit words per second over the PCM channel which is a standard 64 kilobit/sec PCM channel.

A sync signal is generated by the local switching network (not shown) in a known manner and supplied to a voice/data combiner via an input line 31. Preferably, this sync signal has a frequency of 400 hertz and a 15% duty cycle, but other sync signals may be used as required for specific system performance. The sync signal is connected to phase lock loop 1 which locks on to the sync signal and provides various clock references for controlling the voice/data combiner, the encoder 8 and the decoder 9.

When both voice and data information are to be transmitted simultaneously ("voice/data mode"), switches 40, 41, 42, and 43 are opened, thus inserting various system elements in the path between the encoder 8 and switch interface lead 32, and the path

between a switch interface lead 33 and the decoder 9. During the reception of a voice/data signal, the PCM signal is received via the switch interface lead 33. The serial data stream is fed to serial-to-parallel register 12, which provides an output 8 bits wide. A device which may be used as a serial-to-parallel register is the SN74164, manufactured by Texas Instruments, Inc. This parallel output signal from serial-to-parallel register 12 is connected to a demultiplexor 18. Demultiplexor 18 is clocked by a signal made available from the phase lock loop 1, such that during the reception of the data portion of the PCM input signal, the parallel output from register 12 is connected by demultiplexor 18 to receive-data-buffer 15, and during the reception of the voice portion of PCM input signal, the output from register 12 is connected by demultiplexor 18 to receive-voice-buffer 14. For the system described, the sync signal depicted in Figure 2b is high during the first three (3) bytes, which are the data bytes. Thus, this 400 Hz, 15% duty cycle syc signal is used to control multiplexor 7 and demultiplexor 18 such that data information is transmitted and received during the first three (3) bytes of each frame, and analog information is transmitted and received during the remaining seventeen (17) bytes per frame. For a system where three 8-bit data bytes are transmitted per frame, receive-data-buffer 15 is a 3 word by 8 bit memory. Similarly, for a system where 17 8-bit voice bytes are transmitted in a single frame, receive voice buffer 14 is a 17 word by 8 bit memory. For the system employing 17 words per 2.5 msec frame during the voice/data mode, encoder 8 and decoder 9 operate at 6.8 KHz, resulting in a digitized voice rate of 54.4 kilobits/sec.

During the reception of a voice/data signal, each frame has a period of 2.5 mS, as shown in Figure 2a.

Since each frame contains 20 bytes, each byte is transmitted in 125 microseconds. Thus, an 8-bit byte is output from serial-to-parallel register 12 every 125 microseconds. Each data byte is clocked into data buffer 15 at the rate of one byte per 125 microseconds, although only 3 data bytes are input during each frame, as shown in Figure 2c. Similarly, each voice byte is shifted into voice buffer 14 in 125 microseconds, although only 17 voice bytes are shifted per frame. A sync signal (shown in Figure 2b), available from local switching equipment (not shown) has a frequency of 400 Hz and a 15% duty cycle. Thus the sync signal is high during the first three bytes of each frame, which corresponds to the data bytes, thus allowing the multiplexor 7 and the demultiplexor 18 to selectively interface with voice buffers 4 and 14, and data buffers 5 and 15, as required. Multiplexor 7 may comprise two SN74157 devices manufactured, for example, by Texas Instruments, Inc. and demultiplexor 18 may comprise two SN74LS244 devices, manufactured, for example, by Texas Instruments, Inc.

Each voice and data byte comprises 8 bits as shown in Figure 2d. The data is shifted out of data buffer 15 at the rate of approximately 833 microseconds/byte, or three bytes per frame, thereby providing a continuous stream of data output, as shown in Figure 2F. This 833 microseconds/byte clock signal, as well as all other clock signals used to control the various buffers, registers, multiplexors, demultiplexors, encoders and decoders are generated by the phase lock loop 1 in a well-known manner. The phase lock loop uses the sync signal, as depicted in Figure 2b, as a reference, thereby providing accurate clock signals for the operation of the system. This data output is connected through an interface 6 to a data receiver (not shown).

Similarly, voice information is shifted out of voice buffer 14 at approximately 147 microseconds per byte, or 17 butes per frame, thereby providing a continuous stream of voice information to the decoder 9, as shown in Figure 2e. This continuous stream is desired to provide high quality voice transmission. If, on the other hand, the voice sampling rate remains constant, and during the voice/data mode three (3) voice bytes are replaced by three (3) data bytes during each frame, three (3) voice bytes per frame will be lost, resulting in a noticeable degradation in quality of the transmitted voice signal.

In a similar fashion, voice information is received from input lead 30 of the encoder 8, for exampling at appoximately 147 microseconds per byte during the voice/data mode. This information is fed to serial-to-parallel register 2, which converts the serial stream of bits from the encoder 8 to a parallel stream of information, 8 bits wide. The parallel voice information from the register 2 is clocked into transmit voice buffer 4 at approximately 147 microseconds per byte. Voice buffer 4 is a 17 word by 8 bit memory. Data information is received from a data terminal (not shown) through interface 6 to transmit data buffer 5, at approximately 833 microseconds per byte. Transmit data buffer 5 is a 3 word by 8 bit memory. Voice information and data information are clocked out of buffers 4 and 5 at 125 microseconds per byte. During the first 3 bytes per frame, data is clocked out of the data buffer, through multiplexor 7, and to parallel-to-serial register 13. During the remaining 17 bytes per frame, voice information is shifted out of voice buffer 4, through multiplexor 7 to parallel-to-serial register 13. Parallel-to-serial register 13 converts an 8 bit wide byte into a continuous stream of binary data which is connected to switch interface

lead 32 and transmitted over a standard 64 kilobit/sec PCM channel.

This system also includes means for queuing each station in a communications network, such that each station is either in the full voice mode, or the voice/data mode, as required. When in the full voice mode, a unique audible or subaudible tone (or set of tones) is transmitted by a first station to each associated station in communication with the first station, over the same PCM transmission channel used to transmit voice and data information, in order to signal all associated stations to switch to the voice/data mode. With stations in the voice/data mode, the same or a different tone may be transmitted over the PCM channnel by one station to signal all associated stations to switch to the full voice mode. Such a tone, or set of tones, is generated by tone generator 100, of well-known design, as shown in Figure 1, and then applied to pulse code modulator 8. The tone is detected by a tone decoder 101, and a signal applied to the interface and control logic 6, which then causes the voice/data combiner to enter the voice/data mode.

Alternatively, a special binary word, or set of words, representative of a queuing tone, or set of tones, is transmitted by one station to signal a change from the voice/data mode to the full voice mode. These tone signals are stored in PCM form in a read only memory queuing ROM 110 in voice/data combiner and applied to parallel-to-serial register 13 as needed to be transmitted as a voice signal, as controlled by interface and control logic 6. In a similar manner, a unique data word, or set of words, are stored in ROM 110 contained within the voice/data combiner, and transmitted as a data signal as required to signal a transistion from the voice/data mode to the full voice mode. This data signal is decoded by interface and control logic 6, which then

causes the voide/data combiner to enter the voice/data mode.

Alternatively, a special binary word, or set of words, representative of a queuing tone, or set of tones, is transmitted by one station to signal a change from the voice/data mode to the full voice mode. These tone signals are stored in PCM form in a read only memory queuing ROM 110 in voice/data combiner and applied to parallel-to-serial register 13 as needed to be transmitted as a voice signal, as con-rolled by interface and control logic 6. In a similar manner, a unique data word, or set of words, are stored in ROM 110 contained within the voice/data combiner, and transmitted as a data signal as required to signal a transition from the voice/data mode to the full voice mode. This data signal is decoded by interface and control logic 6, which then switches the voice/data combiner to the full voice mode. These alternative binary queuing signals are also transmitted over the same PCM channel used to transmit voice and data information, thus eliminating the need for an additional, independent queuing channel. The binary queuing signals must be selected in a manner that is compatible with the sampling rate of the system. For example, if the binary queuing signal is to be transmitted as a voice signal, it must have the same format (i.e., same number of bits, and the same transmission rate) as voice signals. In a similar manner, if the binary queuing signal is to be transmitted as a data signal, it must have the same format as a data signal.

CLAIMS

1. A method of transmitting a voice signal and data information on a single channel, the method comprising the steps of

(a) sampling the voice signal at a first rate in the absence of data to be transmitted, and

(b) sampling the voice signal at a second rate, lower than the first rate, and sampling the data information at a third rate lower than the first rate, during the presence of data to be transmitted.

2. A method as claimed in claim 1 wherein the combined sampling rate of said voice signal and said data information during the presence of data to be transmitted is equal to the sampling rate of said voice signal during the absence of data to be transmitted.

3. A method of transmitting a voice signal and data information on a single channel, the method comprising the steps of

(a) pulse code modulating the voice signal at a first sampling rate into a series of voice frames during periods when only voice information is to be transmitted, each voice frame comprising a plurality of voice bytes;

(b) pulse code modulating the voice signal at a second sampling rate into a series of voice frames during periods when both voice information and data information is to be transmitted, each voice frame comprising a plurality of voice bytes;

(c) merging one or more data bytes with one or more voice bytes to form a voice/data frame during periods when both voice information and data information are to be transmitted, thereby forming a series of voice/data frames comprising both voice

bytes and data bytes;

(d) transmitting the voice frames to a receiving location during periods when voice information only is to be transmitted;

(e) transmitting the voice/data frames to a receiving location during periods when both voice information and data information are to be transmitted;

(f) separating at the receiving location the voice bytes and the data bytes in each voice/data frame; and

(g) pulse code demodulating at the receiving location the voice bytes to an analog voice signal.

4. An apparatus for transmitting a voice signal and dat a information on a single channel, the apparatus comprising means for sampling the voice signal at a first rate in the absence of data to be transmitted, means for exampling the voice signal at a second rate, lower than the first rate and sampling the data information at a third rate lowerthan the first rate during the presence of data to be transmitted.

5. An apparatus as claimed in claim 4 wherein the combined sampling rate of the voice signal and the data information during the presence of data to be transmitted is equal to the sampling rate of the voice signal during the absence of data to be transmitted.

6. An apparatus for the transmission and reception of analog information and digital data simultaneously, the apparatus comprising:

an analog input terminal for the reception of an analog input signal;

an analog-to-digital converter having an output terminal, for converting said analog input signal to a digital representation thereof, the

analog-to-digital converter being capable of sampling the analog input signal at a first sampling rate during periods when only analog information is to be transmitted, and at a second sampling rate during periods when analog information and digital data are to be transmitted simultaneously;

a pulse code modulation output terminal connected to a transmission channel;

means for connecting the analog-to-digital converter output terminal to the pulse code modulation output terminal during periods when only analog information is to be transmitted;

a pulse code modulation input terminal connected to the transmission channel;

a digital-to-analog converter having an input terminal and an output terminal capable of operating at the first sampling rate during periods when only voice information is to be received and at the second sampling rate during periods when analog information and digital data are to be received simultaneously;

means for connecting the digital-to-analog converter input terminal to the pulse code modulation input terminal during periods when only analog information is to be transmitted;

means for storing the digital representation of the analog signal during periods when both analog information and digital data are to be transmitted;

means for storing the digital data during periods when both analog information and digital data are to be transmitted;

means for selectively outputting the digital representation of the analog signal and the digital data to the pulse code modulation output terminal during periods when both analog

information and digital data are to be transmitted;

means for separating and storing the received digital representation of analog information and the received digital data received at the pulse code modulation input terminal;

means for outputting the received digital data; and

means for applying the received digital representation of analog information to said digital-to-analog converter.

7. An apparatus as claimed in claim 6 having a tone generator connected to the analog input terminal for transmitting a queuing signal indicative of a change from the mode of transmitting only analog information to the mode of transmitting analog information and digital data simultaneously, or vice-versa, and a tone decoder connected to the digital-to-analog converter responsive to the queuing signal for controlling the samping rates of the digital-to-analog converter and of the analog-to-digital converter.

8. An apparatus as claimed in claim 6 having means for transmitting a digital queuing signal indicative of a change from the mode of transmitting only analog information to the mode of transmitting analog information and digital data simultaneously, or vice-versa, and means responsive to the queuing signal for controlling the sampling rates of the digital-to-analog converter and of the analog-to-digital converter.

9. An apparatus for the simultaneous transmission of analog information and digital information over a single transmission channel, the apparatus comprising

an analog to digital converter arranged for operating at a first sampling rate during periods when only analog information is to be

transmitted and at a second sampling rate during periods when both analog information and digital information are to be transmitted, the analog-to-digital converter having an input means for the reception of analog information and an output means for the output of a digital representation thereof comprising a series of voice words each having one or more bits;

a first buffer memory for storing a plurality of one or more voice words;

a second buffer memory for storing a plurality of one or more data words, each data word comprising one or more bits;

means for merging the plurality of voice words stored in the first buffer memory with the plurality of data words stored in the second buffer memory; and

means for transmitting the merged voice and data words.

10. An apparatus for the simultaneous reception of analog information and digital information from a pulse code modulated transmission of a series of words each having one or more bits, the apparatus comprising

means for storing in a first buffer memory the words corresponding to the analog information;

output means for the words corresponding to the digital information; and

digital to analog converter means for converting the words stored in said first buffer memory to an analog signal, the digital-to-analog converter means being capable of operating at a first sampling rate during periods when only analog information is to be received, and at a second sampling rate during periods when both analog information and digital

information are to be received.

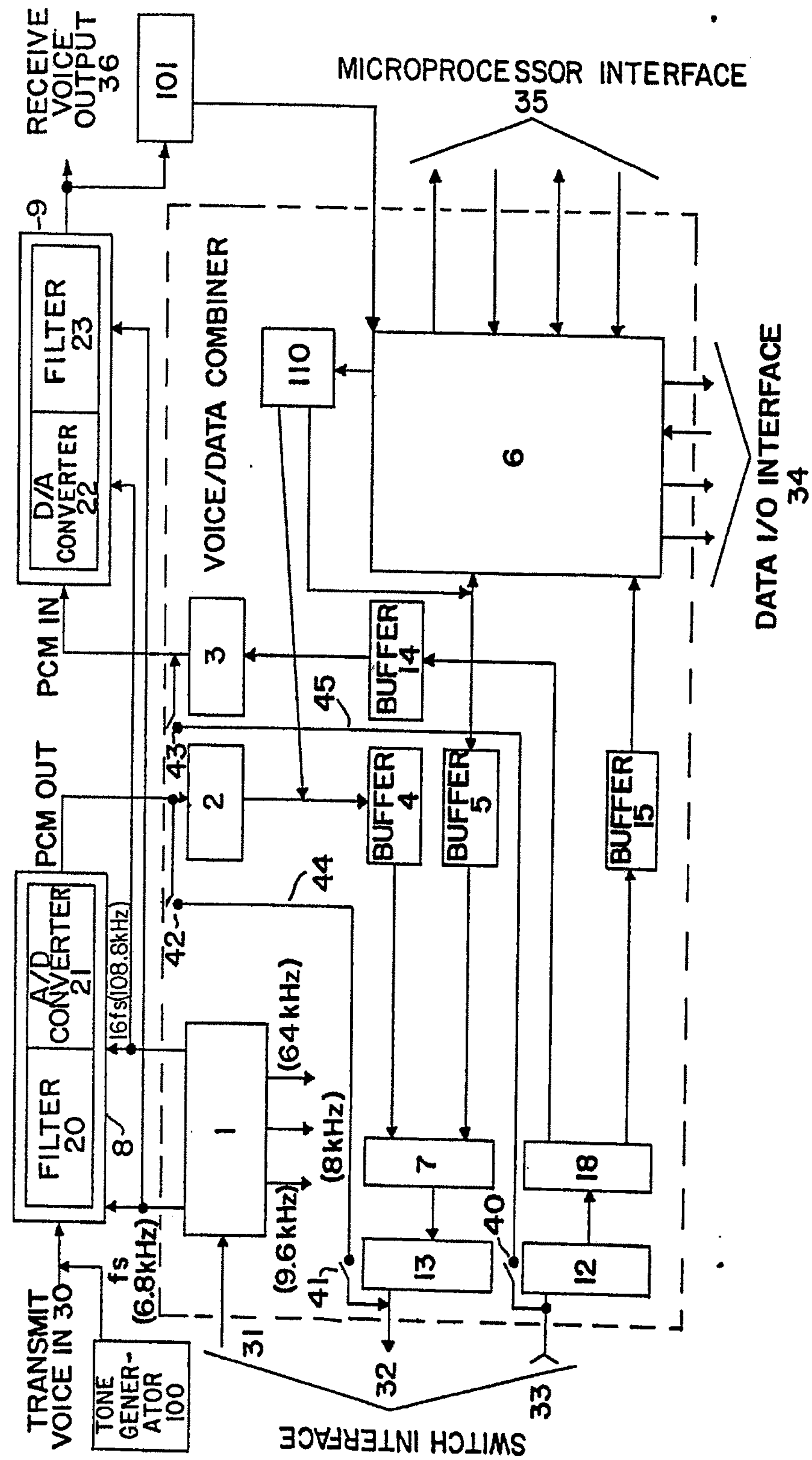
RECEIVE VOICE OUTPUT 36
101
MICROPROCESSOR INTERFACE 35
9
FILTER 23
D/A CONVERTER 22
VOICE/DATA COMBINER
110
6
DATA I/O INTERFACE 34
PCM IN
3
BUFFER 14
45
PCM OUT
43
2
BUFFER 4
BUFFER 5
BUFFER 15
44
42
A/D CONVERTER 21
16fs(108.8kHz)
(64 kHz)
FILTER 20
8
1
(8kHz)
(9.6 kHz)
7
18
40
41
13
12
fs (6.8kHz)
TRANSMIT VOICE IN 30
TONE GENERATOR 100
31
32
33
SWITCH INTERFACE

FIG. I

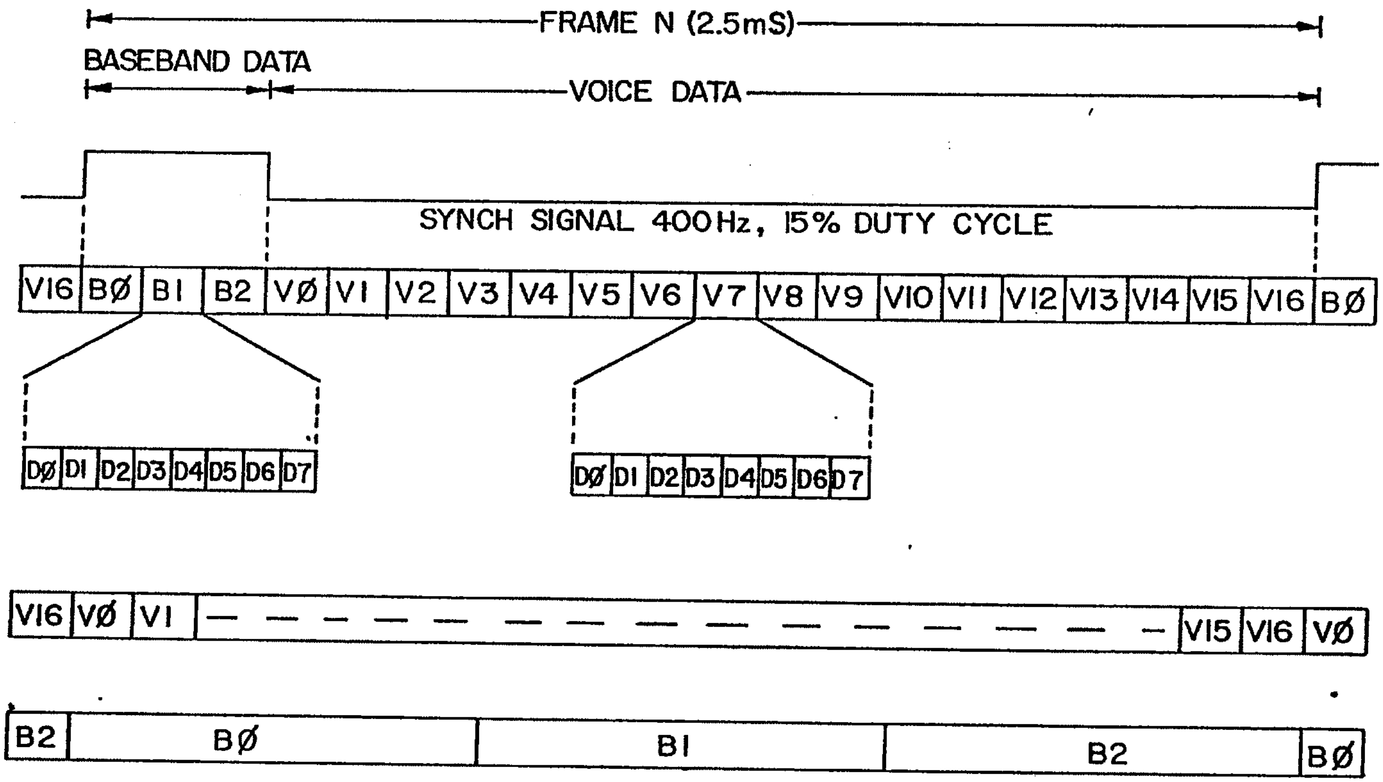

FIG. 2a

FIG. 2b

FIG. 2c

FIG. 2d

FIG. 2e

FIG. 2f